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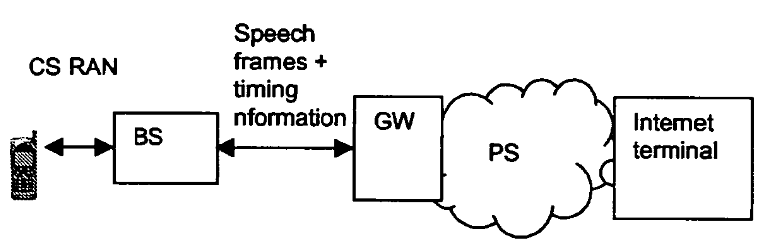
(54) Transmission timing for data packets transmitted from an internet terminal to a terminal in a cellular radio network

(57) A method for transferring data over a communication link from a first unit (e.g. an Internet terminal) to a second unit (e.g. alterminal in a cellular radio network), the link comprising a first portion across which the data is carried by the transmission of data packets at regular intervals (e.g. the radio network), and a second portion between the first portion and the first unit over which the data is carried in a form that is not synchronised with the transmission of data packets over the first portion (e.g. a packet switched network such as an IP network); the method comprising transmitting to the first unit time alignment information regarding the times at which packets are to be transmitted over the first portion.

Figure 3



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Description

[0001] This invention relates to controlling transmission timing, and especially remote controlling of transmission timing in packet switched media communication systems.

[0002] Practically all modern telephony applications use speech compression to increase the efficiency with which the transmission media are used. The functional entity that performs the compression is called a speech codec. Most of the modern speech codecs operate by processing the speech signal in short segments called frames. For instance, all GSM (global system for mobile communications) codecs, including the AMR (adaptive multi-rate) codec, use 20 ms frames.

[0003] One commonly known property of a telephone link is that it is very sensitive to the delay introduced by the transmission of speech data transmission time from sender to receiver and back (so called round trip delay). Practical tests have shown that even relatively short round trip delay (around 400ms) degrades the interactivity of the discussion, and round trip delays over 800ms are found to reduce the quality of Service (QoS) to an unacceptable level. Therefore, generally a telephony system should be designed in such a way that the maximum round trip delay can be limited below a predetermined threshold so as to provide predictable and acceptable quality.

[0004] Traditional telephony services use the circuit switched (CS) approach. This means that the parties to the connection communicate over a transmission channel that is reserved for the whole duration of the communication. This implies that the data is transmitted over a fixed route, and also the transmission time is fixed and predictable. Therefore, this kind of telephone network can offer reliable service with controlled QoS. An important group of applications employing CS telephone services are some cellular mobile systems, e.g. GSM.

[0005] On the other hand, the emergence of the Internet has created a new platform for telephony applications: There are already a number of telephony applications which use packet switched (PS) networks (such as the Internet) to transmit speech data. Most, although not all, PS networks are based on IP (internet protocol) protocols (like the Internet) and telephony applications running on this kind of networks are referred as IP telephony or Voice-over-IP (VoIP). The basic idea of a PS network is that the transmitted data is decomposed into small sub-blocks called packets, and the receiving application uses received packets to recompose the original data. Each packet can be transmitted from source to destination independently of other packets, and it is up to the network to route packets from source to destination. This implies that it is quite possible that the packets belonging to the same stream will use different routes to reach the destination. Furthermore, in general a PS network provides only a so-called 'best effort' service: the packets are transmitted from source to destination without any guarantees about the QoS. Therefore, it is possible that some of the packets are lost during transmission, and the time required for the transmission from source to destination is in the general case unpredictable. Due to varying load in the network and possibly also to different transmission paths of the packets, the transmission delay can vary from packet to packet within a stream. This variation in transmission time is called jitter. Considering the Internet in general, the transmission delay can vary from a negligible level to even several seconds. The same applies also to jitter, although usually the transmission time and jitter are related: in many cases long transmission time also means large jitter. This unpredictable delay behaviour is likely to cause quality problems for VoIP services. However, in a relatively small and closed IP network, such as a company LAN (Local Area Network), the delay and jitter can often be limited to a desired range by network design and by controlling the amount of traffic that is allowed into the network.

[0006] As an example, in the current GSM system the CS approach has been extended to cover data services over a CS radio channel. Because of the narrow bandwidth offered by the radio system (which was originally designed for speech services), the data rates offered are relatively low. In spite of this, these services have gained popularity, and rapid advances in radio technology are expected to significantly increase available data rates in the near future. On the other hand, the Internet offers a vast range of services, and therefore it would be appealing to combine these 'two worlds' to extend the coverage of the 'Internet services' also to mobile users. The convergence is also appealing from the telephony point of view, the scenario being that of a connection between a terminal in a cellular mobile (radio) network and a terminal in a VoIP domain.

[0007] One proposed system would include both CS and PS radio access networks (RANs), together with a PS core network (CN). Furthermore, the CN part of the network could be connected to an external PS network (such as the Internet or a company LAN) through a gateway (GW), thus enabling a connection to a terminal connected to this external network via its own access network (AN). This could conceivably enable seamless and transparent connection between terminals anywhere within reach of a concatenation of networks. Figure 1 presents a greatly simplified illustration of this arrangement.

[0008] In a PS network, speech frames are typically transmitted using the Real-time Transport Protocol (RTP) packets. (See IETF RFC 1889 "RTP: A Transport Protocol for Real-Time Applications", 1996). Furthermore, RTP is typically run over User Datagram Protocol (UDP) and IP. (See IETF RFC 768 "User Datagram Protocol", 1980). GSM speech frames can be encapsulated into RTP packets according to the standard specified in ETSI TS 101 318 "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Using GSM speech codecs within ITU-T

Recommendation H.323", v1.1.1, 1998. Currently, the IETF is also working on specifying a method to encapsulate AMR speech frames into RTP. This will be an important specification for 3G work, since the AMR codec has been selected to be the only mandatory speech codec for 3G systems.

[0009] The RTP Control Protocol (RTCP) is an integral part of the RTP specification. Whenever RTP packets are used, RTCP packets should also be transmitted. (See IETF RFC 1889 "RTP: A Transport Protocol for Real-Time Applications", 1996). RTCP is used to monitor quality of service and to give information about the participants. RTCP packets are transmitted periodically, typically less often than RTP packets to save bandwidth (see section 6.2 of the IETF RFC 1889).

[0010] In the communication situation described above (and illustrated in figure 1), radio bandwidth is arguably the most scarce resource on a path from a fixed VoIP terminal to a mobile terminal in a cellular network. Furthermore, transmission over a RAN is likely to introduce a considerable amount of delay. Therefore, the radio link can be regarded as the 'bottleneck' within this connection, and it would be advantageous to try to optimise the use of radio band.

[0011] The efficient use of radio bandwidth requires strict scheduling of transmitted data, and this usually means that radio frames must be transmitted at fixed intervals. Furthermore, efficient radio transmission usually also implies that the data from different sources ('logical channels') is transmitted on the same radio block ('physical channel'). In pure CS environments this normally does not have any effect on the performance/delay of the system. On the other hand, the entity controlling the radio transmission timing does not have any control over transmission times of a terminal that is located in the PS VoIP domain. Transmission over the external PS domain is asynchronous, and in this kind of case the frames from different sources scheduled for radio transmission in the same radio block arrive at the RAN at different times and have to be buffered to wait for further transmission over the radio link.

[0012] Figure 2 shows schematically the arrangement of a GSM mobile station , BTS (Base Transceiver Station) and BSC (Base Station Controller). The GSM mobile is connected to via radio interface to a BTS. Speech frames are transmitted between BTS and BSC in TRAU (Transcoder/Rate Adaptor Unit) frames. Speech frames are encoded/decoded in the TRAU unit, which is typically located in the BSC. Delay between GSM mobile and BSC may change during a call, since:

1. the time slot may change,
2. the GSM mobile may change from one BTS to another BTS inside the BSC area. Normally, TRAU frames are transmitted every 20 ms. However, it is possible to change the length of the TRAU frames (and thus the transmission period) by changing the number of stop-bits located at the end of the TRAU frame.

[0013] To handle uplink timing, the BTS sends TRAU frames when those are received from the radio channel. The TRAU unit located in the BSC decodes the TRAU frames to speech samples, which are sent to the PCM line. Since the sampling interval is fixed in the PCM line, the TRAU unit can skip or repeat speech samples to adjust the timing in case the arrival of a TRAU frame differs from the nominal frame length 20 ms.

[0014] To handle downlink timing, the BTS sends TRAU frames to the radio channel at fixed intervals depending on timing in the radio channel. At the beginning of the call BSC has no information about timing at the BTS. Additionally, if the time slot or the BTS changes, the optimal timing changes too. To adjust the timing, BTS sends timing information to the BSC. According to this time alignment information, the BSC adjusts transmission time of the downlink TRAU frames. Again, transmission time can be adjusted by repeating or skipping PCM speech samples.

[0015] The above mentioned timing method is explained in detail in GSM 08.60 "Digital cellular telecommunications system (Phase 2+); In-band control of remote transcoders and rate adaptors for full rate traffic channels", v8.1.0, 1999 at chapter 4.6.1 "Time Alignment of the speech service frames".

[0016] According to the present invention there is provided a method for transferring data over a communication link from a first unit to a second unit, the link comprising a first portion across which the data is carried by the transmission of data packets at regular intervals, and a second portion between the first portion and the first unit over which the data is carried in a form that is not synchronised with the transmission of data packets over the first portion; the method comprising transmitting to the first unit synchronisation information regarding the times at which packets are to be transmitted over the first portion.

[0017] Preferred features of the invention are set out in the dependant claims.

[0018] The present invention will now be described by way of example with reference to the accompanying drawings, in which:

figure 1 shows a Connection between a mobile terminal and a fixed VoIP terminal;
figure 2 illustrates speech transmission in a GSM system;
figure 3 illustrates a call via CS RAN to an internet terminal.

[0019] An embodiment of the invention will be described with reference to telephony applications. However, similar

principles could be applied to any appropriate delay critical applications in which frame based media are used with fixed transmission intervals between successive data frames within a stream.

[0020] The PS network may introduce quite a long transmission delay and large jitter. In this case, proportional advance of a timing control will be quite small. However, an interactive packet based speech service is of limited attraction unless the delay can be limited to a small enough value. Thus, the VoIP domain should comprise a well regulated network with small jitter and relatively low transmission delay. In this case synchronization with the RAN can afford savings in overall delay. In many cases a LAN can provide small enough jitter and delay. Also a normal internet connection may fulfil these requirements, especially if QoS classifications are used.

[0021] Since the system might be operating near the maximum allowable delay, even small savings in the overall delay can make the system more feasible. Therefore, in this embodiment a method for controlling the transmission scheduling of a remote VoIP terminal (remote terminal is a terminal that is located in a different network than the other party of the connection) is provided.

[0022] In this embodiment the entity controlling transmission over a synchronized link (in this example a radio link) can send requests to the remote terminal with which it has established a link to adjust its transmission timing to match that of the radio link. The adjustment of transmission time can be performed at call set up and/or during the call:

1. The time adjustment can be performed during the call set-up (when the connection is being established) by indicating the correct 'grid' of the transmission times. The initial control might require synchronization of the clocks in both ends. This can be performed e.g. by using Network Time Protocol (NTP). (See IETF RFC 1305 "Network Time Protocol (Version 3): Specification, Implementation and Analysis", 1992). The timing adjustment information can be sent using a proprietary protocol, or it could be possible to specify e.g. an RTCP message subtype for this purpose.

2. During a call there may be a need for re-adjustment due to, for example, handover in the radio network or to clock drift in the remote terminal. In case of on-line timing adjustment it is also desirable to have the capacity to manipulate the transmitted media by dropping data or generating some extra data, if the transmission time 'grid' is changed forward or backwards. In the case of a speech application the change could be performed safely with the aid of the speech encoder: the adjustment could be performed during a speech pause or during a period of speech in which it is determined that the manipulation of the signal is unlikely to have a great impact on speech quality.

[0023] By synchronisation of the data transmission with the transmission slots available in the network overall transmission delay can be reduced. In theory the amount of saving in delay is up to the block duration of the transmitted media. For example with the AMR codec using 20ms frames this method can save up to 20ms in one-way transmission time. If the transmission is configured to encapsulate several consecutive frames into each transmitted packet, in some scenarios the possible saving could be even bigger.

[0024] One way to provide for the synchronisation information to be carried is by extending the RTCP protocol, for example by adding two optional fields to the RTCP packet. These fields contain an identifier and data as follows:

Identifier	TIME_ALIGNMENT_REQUEST (a constant)
Data	16 bit signed integer indicating the amount of time shift to be done. The unit indication the change is the same as the unit of the timestamp in the RTP packet

Identifier	TIME_ALIGNMENT_RESPONSE (a constant)
Data	16 bit signed integer, which indicates the how much the timing has been changed. Typically, this is the same as in the request. However, in same cases the timing change may be implemented in small steps and further response messages will be transmitted later. The value may also be different if the time alignment can not be implemented with the asked resolution (in this case the closest value is chosen). Value "0" indicates that this entity is not capable of changing the timing.

[0025] In use, the receiving unit sends a TIME_ALIGNMENT_REQUEST message to ask the transmitting unit to change the timing. When the transmitting unit has changed its timing, it indicates the change by sending TIME_ALIGNMENT_RESPONSE.

[0026] One example application of the system will be described with reference to figure 3.

[0027] Coded speech is transmitted over a CS RAN and through a GW to an internet terminal. As explained earlier,

the packet transmission time of the CS RAN is fixed; in other words speech frames can be sent and received at fixed moments of time. The timing may also change during a call. The internet terminal, which contains also processing means for implementing the speech codec, can freely change its timing.

[0028] At the beginning of the call, the internet terminal starts to send the packets at an arbitrary moment. The GW has to buffer the incoming packets to fit them with CS RAN timing (the GW and BS (Base Station) exchange timing information). To minimize the delay, GW and the internet terminal can use the time alignment according to the following steps. For instance, if the AMR (Adaptive Multi-Rate) codec using 20ms frames is employed, and the internet terminal transmits data with one frame per packet, and we assume that there is no jitter, the packets (frames) arrive at the GW at 20ms intervals:

$$t_r(n) = t_0 + n^*20,$$

where $t_r(n)$ indicates the reception time at GW for packet n, and t_0 indicates the time when the first packet of the stream was received. Now if the synchronized radio link (in BS) requires that the packets are transmitted to the radio path at

$$t_t(m) = t_0 + m^*20 +18,$$

every packet would have to wait 18ms in a buffer in the GW for the next 'transmission slot' on the radio link.

[0029] However, if we use the method proposed in this invention to send a message to the internet terminal and ask it to adjust its transmission in such a way that packets (frames) are sent 18 ms later, the reception from the internet terminal and transmission over the radio link would be 'synchronised', resulting in an 18ms reduction in end-to-end transmission delay.

[0030] The transmission from the internet terminal can be started at an arbitrary moment of time, and the adjustment of transmission timing can be made immediately when the radio link timing is known. Re-adjustment can be needed e.g. because of the handover in the radio network, or because of the clock drift in the internet terminal.

[0031] The GW will typically request a time alignment that is shorter than the frame length. Sometimes it may happen that GW wants to change the timing more than the frame length, for instance in the situation where the GW has many frames in its input buffer. It this case, one option is for the GW simply to remove one or more frames from its buffer. However, this may cause audible deterioration. Instead, a time alignment request can be sent to the speech codec of the internet terminal. In the speech codec, the encoder can remove speech frames that are not important for good speech quality (e.g. frames containing speech pauses).

[0032] In the future, 3G systems or PS internet systems may provide services for conference calls. For voice, this service may contain a unit, which decodes each speech channel, sums the speech signals, encodes the summed signal and sends it back to a participant of the conference call. If the participants send speech packets that are not synchronous, the negotiation unit must delay the packets to be able to sum them. By synchronising the packets to the transmission slots the previously mentioned delay can be reduced.

[0033] The systems described above can be applied to a wide range of networks in which data is transmitted in periodic transmission slots, for example GSM, GERAN, UTRAN, or other types of network.

[0034] Thus by exchanging time alignment information in a call in a PS (packet switched) 'network, delay can be reduced. This is of particular value for real-time applications such as the transmission of speech, audio and/or video.

[0035] The method presented in this invention report can be applied to any delay critical packet based media (speech/ audio/video) that employs constant transmission interval between successive packets within a stream. Examples of application areas are Voice over IP (VoIP) and teleconferencing applications over packet switched networks, but naturally this idea can applied to any delay critical packet based application.

[0036] The applicant draws attention to the fact that the present invention may include any feature or combination of features disclosed herein either implicitly or explicitly or any generalisation thereof, without limitation to the scope of any definitions set out above. In view of the foregoing description it will be evident to a person skilled in the art that various modifications may be made within the scope of the invention.

Claims

1. A method for transferring data over a communication link from a first unit to a second unit, the link comprising a first portion across which the data is carried by the transmission of data packets at regular intervals, and a second portion between the first portion and the first unit over which the data is carried in a form that is not synchronised with the transmission of data packets over the first portion; the method comprising transmitting to the first unit

synchronisation information regarding the times at which packets are to be transmitted over the first portion.

2. A method as claimed in claim 1, comprising the step of the first unit determining when to transmit data towards the second unit based on the synchronisation information.

3. A method as claimed in claim 2, wherein the first unit transmits data towards the second unit as data packets.

4. A method as claimed in claim 3, wherein the first unit transmits the packets towards the second unit at times determined based on the synchronisation information to achieve a lower average delay in the packets entering the first portion of the link than half the period between transmission of packets over the first portion.

5. A method as claimed in any preceding claim, wherein the synchronisation information is sent in the form of an RTCP message.

6. A method as claimed in any preceding claim, wherein the synchronisation information is sent by means of the Network Time Protocol.

7. A method as claimed in any preceding claim, wherein the data is delay-critical data.

8. A method as claimed in claim 7, wherein the data represents speech information.

9. A method as claimed in claim 8, wherein the data is carried is voice-over-IP data.

10. A method for transferring data over a communication link substantially as herein described with reference to the accompanying drawings.

Figure 1

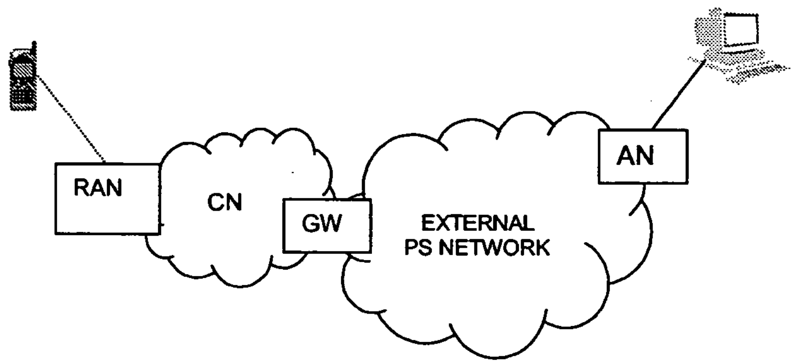


Figure 2

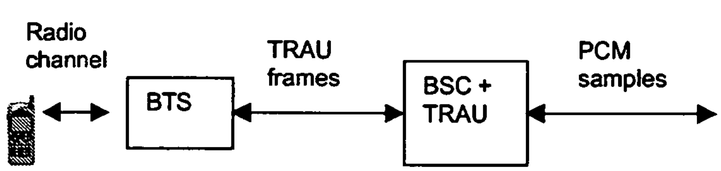
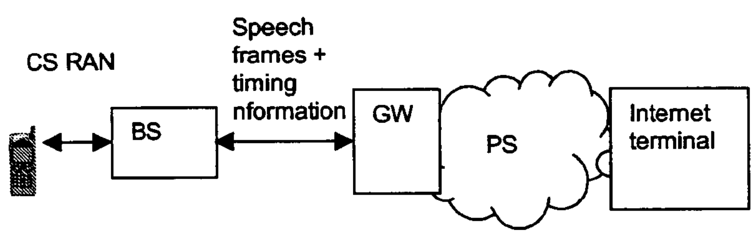


Figure 3


CS RAN
Speech frames + timing nformation
BS
GW
PS
Internet terminal

European Patent Office

PARTIAL EUROPEAN SEARCH REPORT

which under Rule 45 of the European Patent Convention shall be considered, for the purposes of subsequent proceedings, as the European search report

Application Number
EP 02 10 2689

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
X Y	WO 01 15473 A (ERICSSON TELEFON AB L M) 1 March 2001 (2001-03-01) * page 10, line 10 - page 13, line 10 * * page 16, line 13 - page 22, line 17 * * page 32; claim 17 * * figures 1,6 * ---	1-4,7,8 5,6,9	H04Q7/30 H04Q7/22
X Y	WO 01 58097 A (ERICSSON TELEFON AB L M) 9 August 2001 (2001-08-09) * page 7, line 9,10 * * page 8, line 3-11 * * page 12, line 19 - page 13, line 16 * * figure 1 * ---	1-4,7,8 5,6,9	
Y	WO 01 45354 A (NOKIA NETWORKS OY ;KOISTINEN TOMMI (FI)) 21 June 2001 (2001-06-21) * page 4, line 27-36 * * page 6, line 28-31 * * page 2, line 14-27 * --- -/--	5,6,9	TECHNICAL FIELDS SEARCHED (Int.Cl.7) H04Q

INCOMPLETE SEARCH

The Search Division considers that the present application, or one or more of its claims, does/do not comply with the EPC to such an extent that a meaningful search into the state of the art cannot be carried out, or can only be carried out partially, for these claims.

Claims searched completely :
1-9

Claims searched incompletely :

Claims not searched :
10

Reason for the limitation of the search:

Claim 10 relies exclusively in respect of the technical features of the invention on references to the drawings and does therefore not meet the requirements of Article 84 EPC and Rule 29(6) EPC.

Place of search	Date of completion of the search	Examiner
MUNICH	17 March 2003	Möll, H-P

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03 82 (P04C07)

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PARTIAL EUROPEAN SEARCH REPORT

Application Number
EP 02 10 2689

DOCUMENTS CONSIDERED TO BE RELEVANT			CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	
D,A	"DIGITAL CELLULAR TELECOMMUNICATIONS SYSTEM (PHASE 2+);IN-BAND CONTROL OF REMOTE TRANSCODERS AND RATE ADAPTORS FOR ENHANCED FULL RATE (EFR) AND FULL RATE TRAFFIC CHANNELS(GSM 08.60 VERSION 7.0.1 RELEASE 1998), EUROPEAN TELECOMMUNICATIONS STANDARDS INSTITUTE" ETSI EN 300 737 V7.0.1, XX, XX, January 2000 (2000-01), pages 1-31, XP002171819 * page 20, paragraph 4.6.1 - page 21, paragraph 4.6.1.2 * -----	1-9	
			TECHNICAL FIELDS SEARCHED (Int.Cl.7)

EPO FORM 1503 03.82 (P04C10)

ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 02 10 2689

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
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17-03-2003

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EPO FORM P0459

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